US009124693B2

(12) **United States Patent**
Efrati et al.

(10) **Patent No.: US 9,124,693 B2**
(45) **Date of Patent: Sep. 1, 2015**

(54) **METHOD AND SYSTEM FOR ON-HOLD MESSAGING FOR OFF NETWORK CALLS**

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(*) Notice: Subject to any disclaimer, the term of this patent is extended or adjusted under 35 U.S.C. 154(b) by 0 days.

(21) Appl. No.: **14/019,873**

(22) Filed: **Sep. 6, 2013**

(65) **Prior Publication Data**

US 2015/0071124 A1 Mar. 12, 2015

(51) **Int. Cl.**

H04M 7/12	(2006.01)
H04M 3/428	(2006.01)
H04W 4/16	(2009.01)
H04L 29/06	(2006.01)

(52) **U.S. Cl.**

CPC ***H04M 3/4283*** (2013.01); ***H04L 65/1096*** (2013.01); ***H04M 3/4281*** (2013.01); ***H04M 3/4285*** (2013.01); ***H04M 7/12*** (2013.01); ***H04W 4/16*** (2013.01)

(58) **Field of Classification Search**

CPC H04M 3/4288; H04M 3/4285; H04L 65/096; H04L 5/00; H04W 4/16

USPC 379/215.01, 90.01; 370/259; 455/414.1

See application file for complete search history.

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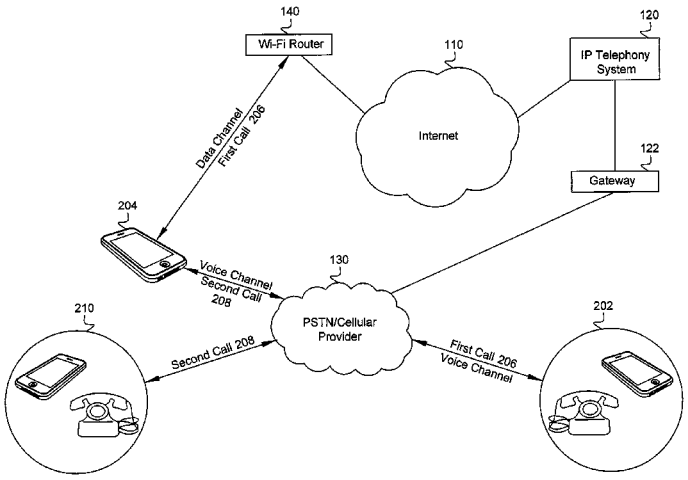
(57) **ABSTRACT**

An Off Net scenario may occur during primary communications between a first and second caller, in which the second caller is connected to a network over a data channel and receives a second call over a voice channel resulting in an interruption of the primary communications between the original callers. In the Off Net scenario, a message indicating the original communication has been interrupted is received by a carrier. The carrier substitutes another communication device in place of the second caller's communication device to establish a secondary communication between the first caller's communication device and the third communication device. The secondary communication may inform the first caller of the interruption of the primary communication and may provide options regarding the primary communication to the first caller.

8 Claims, 6 Drawing Sheets



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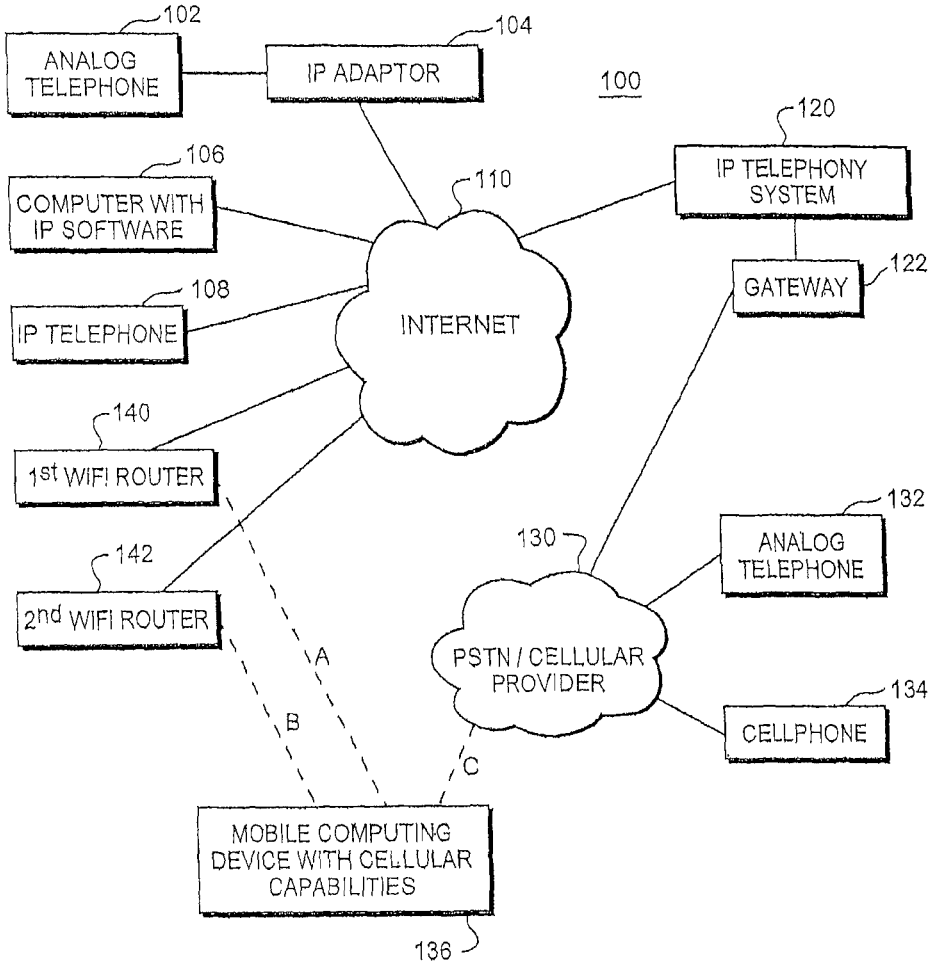
102
ANALOG TELEPHONE
104
IP ADAPTOR
100
120
IP TELEPHONY SYSTEM
106
COMPUTER WITH IP SOFTWARE
110
INTERNET
108
IP TELEPHONE
GATEWAY
122
140
1st WIFI ROUTER
132
ANALOG TELEPHONE
130
142
2nd WIFI ROUTER
PSTN / CELLULAR PROVIDER
A
B
C
134
CELLPHONE
MOBILE COMPUTING DEVICE WITH CELLULAR CAPABILITIES
136

FIGURE 1

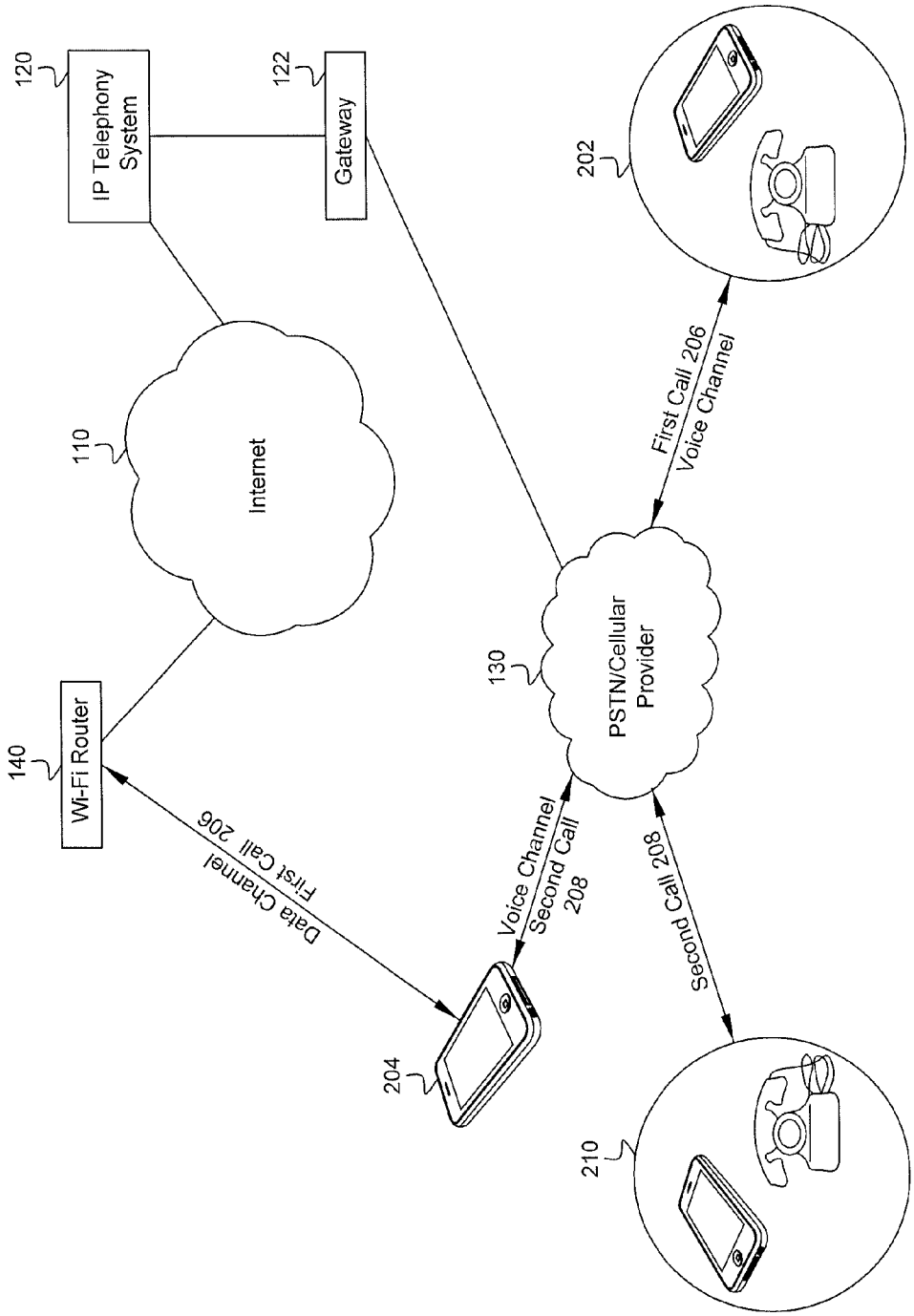
120
IP Telephony System
122
Gateway
202
110
Internet
First Call 206
Voice Channel
130
PSTN/Cellular Provider
140
Wi-Fi Router
Data Channel
First Call 206
Voice Channel
Second Call
208
Second Call 208
204
210

Figure 2

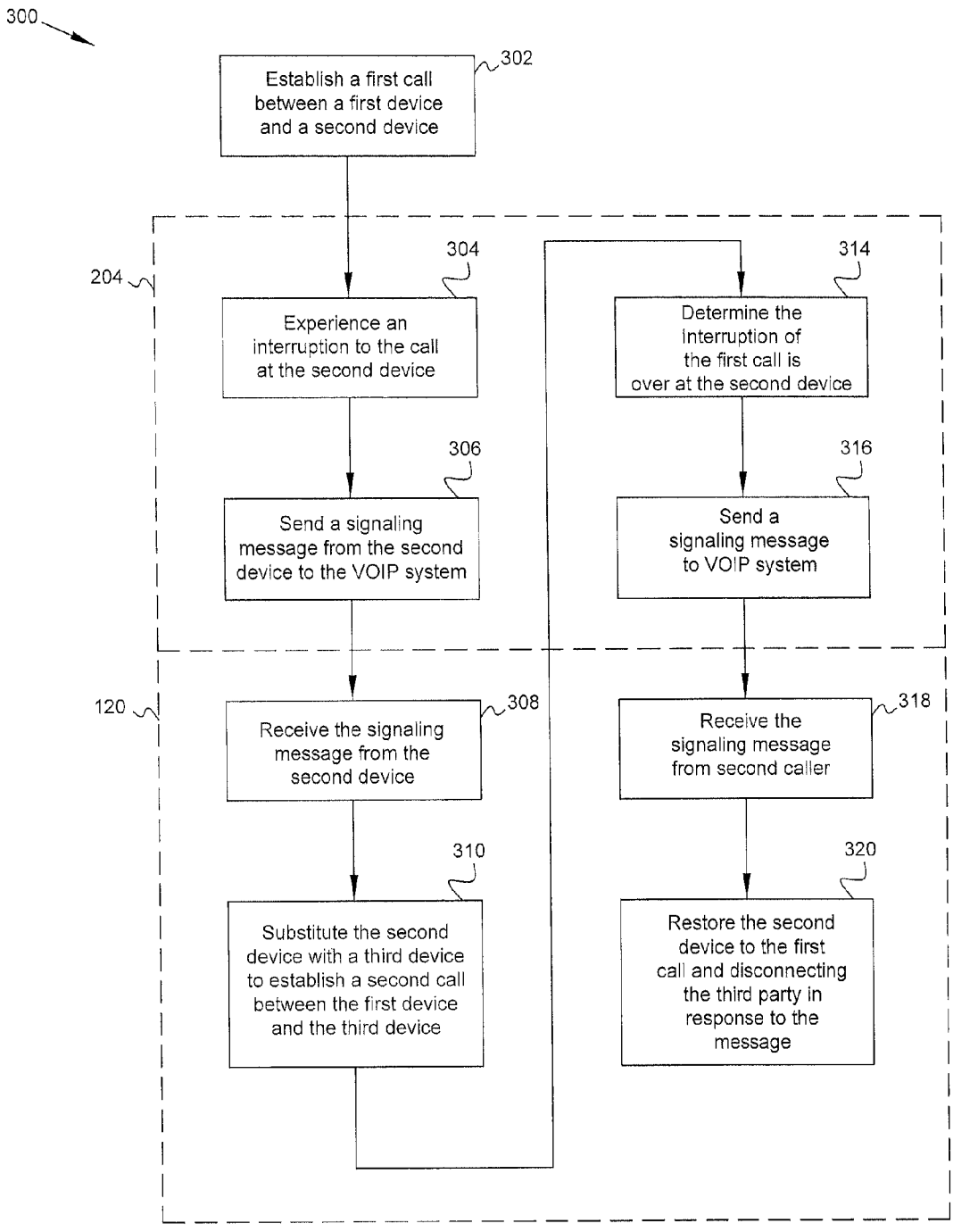
300
Establish a first call between a first device and a second device
302
204
304
Experience an interruption to the call at the second device
306
Send a signaling message from the second device to the VOIP system
314
Determine the interruption of the first call is over at the second device
316
Send a signaling message to VOIP system
120
308
Receive the signaling message from the second device
310
Substitute the second device with a third device to establish a second call between the first device and the third device
318
Receive the signaling message from second caller
320
Restore the second device to the first call and disconnecting the third party in response to the message

Figure 3

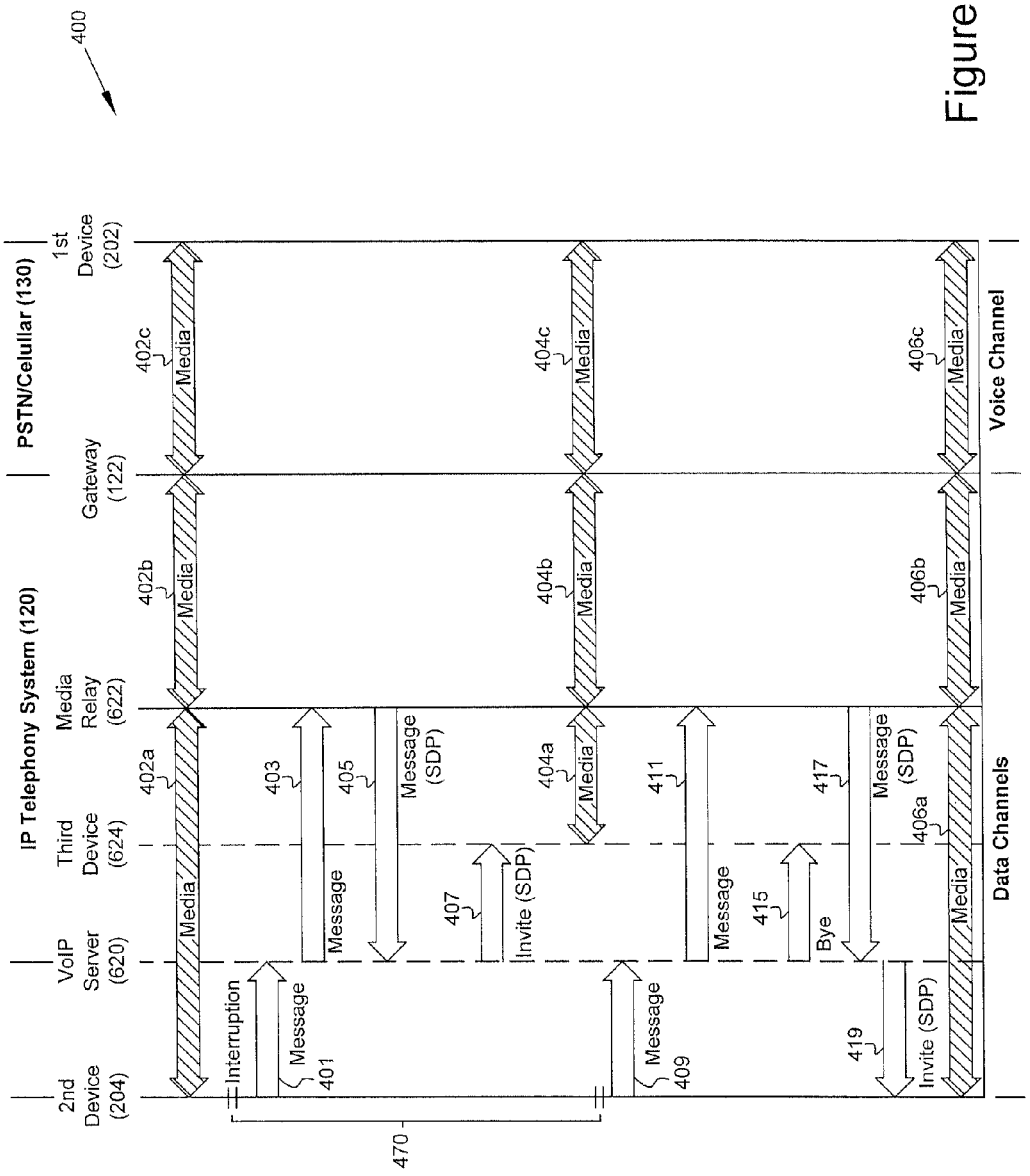
400
IP Telephony System (120)
PSTN/Celullar (130)
2nd Device (204)
VoIP Server (620)
Third Device (624)
Media Relay (622)
Gateway (122)
1st Device (202)
Data Channels
Voice Channel
Interruption
Message 401
403
Message
405
Message (SDP)
407
Invite (SDP)
404a
Media
411
Message
409
Message
415
Bye
417
Message (SDP)
406a
419
Invite (SDP)
Media
402a
402b
402c
404b
404c
406b
406c
470

Figure 4

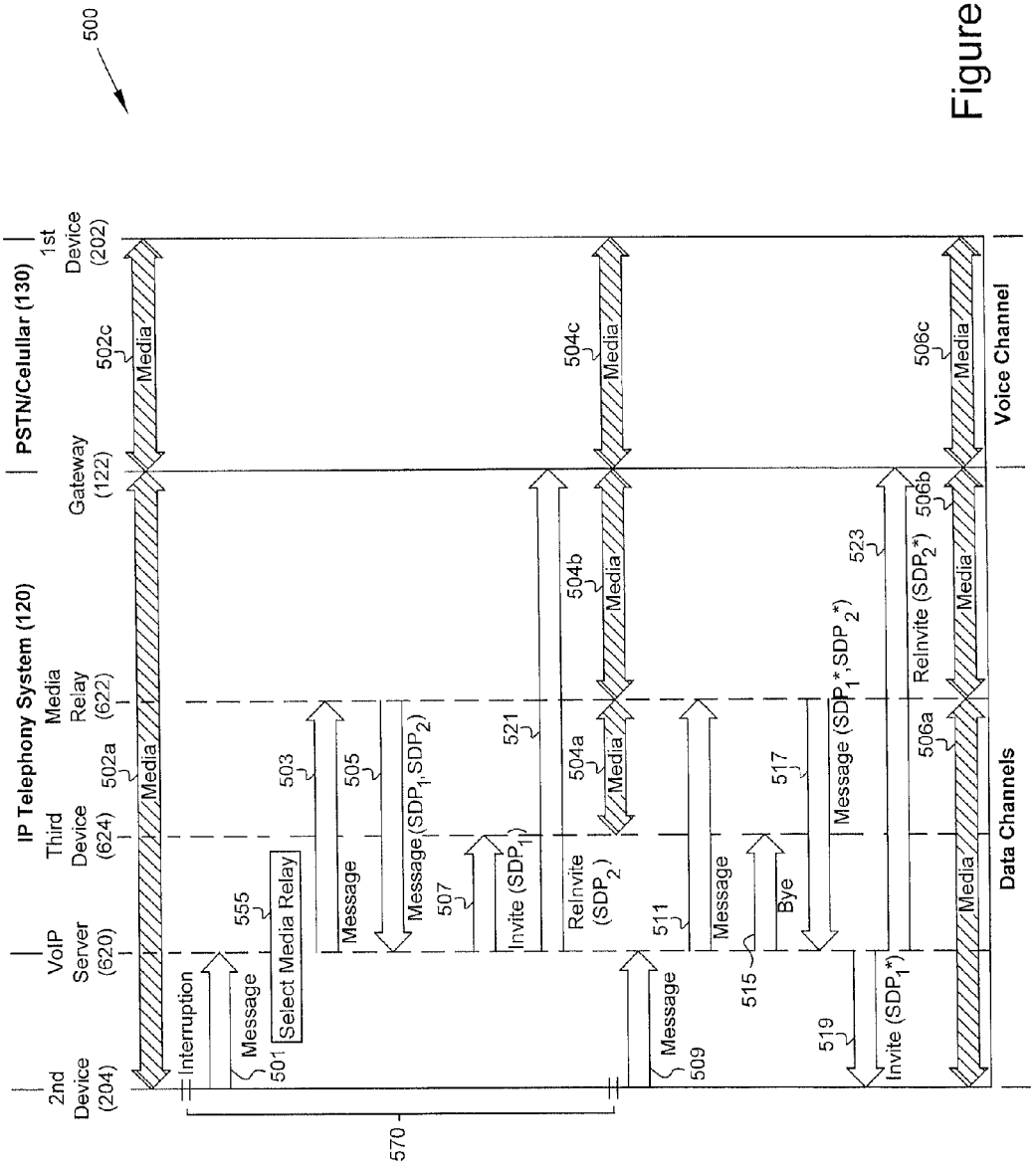
500
IP Telephony System (120)
PSTN/Celullar (130)
2nd Device (204)
VoIP Server (620)
Third Device (624)
Media Relay (622)
Gateway (122)
1st Device (202)
502a
502c
Media
Interruption
Message
501
555
Select Media Relay
503
Message
505
Message (SDP1,SDP2)
507
Invite (SDP1)
521
504a
504b
504c
Media
ReInvite (SDP2)
511
Message
509
Message
515
Bye
517
519
Message (SDP1*,SDP2*)
523
506a
506b
506c
ReInvite (SDP2*)
Invite (SDP1*)
570
Data Channels
Voice Channel

Figure 5

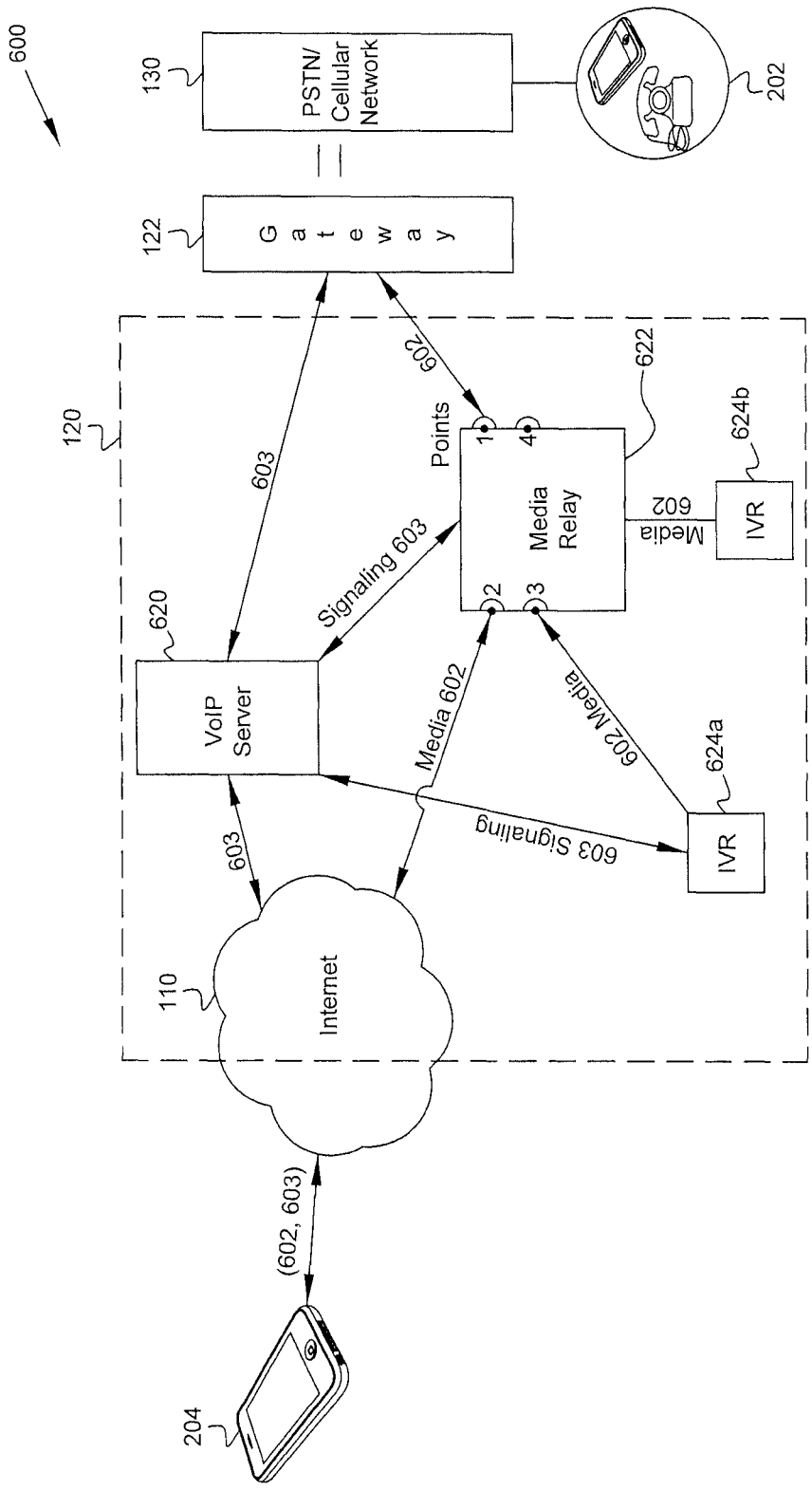
600
130
PSTN/ Cellular Network
202
122
Gateway
120
602
622
Points
1
4
603
624b
Media Relay
Media 602
IVR
Signaling 603
620
2
3
VoIP Server
Media 602
602 Media
624a
603 Signaling
IVR
603
110
Internet
(602, 603)
204

Figure 6

METHOD AND SYSTEM FOR ON-HOLD MESSAGING FOR OFF NETWORK CALLS

BACKGROUND

For many years, people have communicated over traditional landline phones on a PSTN connection and, more recently, over cellular telephones. Increasingly, smartphones are being used over cellular networks providing, among other things, the ability to communicate both voice and data, and allows users to perform a wide range of computing tasks through the availability of mobile "apps," which may be provided by third-party developers.

Smartphones provide for both increased computing features and data network access, and developers provide apps that allow voice communication over the data channel, i.e., voice-over-IP (VoIP). Because these apps use the data channel instead of the voice channel, certain savings may be realized.

However, the use of VoIP apps in place of the voice channel communication that is provided native to the smartphone introduces some challenges. One example is the ability to provide an on-hold message to the other caller. For calls where both callers are on the PSTN or cellular voice channel, if the first caller receives another call and wants to put the second caller on hold, a signal (such as one sent over the SS7 network) may be sent from the second caller's device to the first caller's device instructing it to play an on-hold indicator tone locally. Similarly, where both callers have the VoIP app, conveying on-hold status is readily done; the VoIP provider can provide a signaling message over the VoIP network conveying the on-hold status. However, this does not work if the second caller is communicating over a VoIP app on the data channel and the first caller is on PSTN or cellular voice channel. This is because the second caller's mobile carrier does not recognize that the second caller is even on a call (since it's not using the voice channel and did not otherwise set up a call on the voice channel); and the first caller's carrier may not be equipped to recognize a proprietary on-hold signal from the second caller's VoIP provider. Usually, the voice call over the voice channel will take priority over any other running applications on the second caller's phone, including the VoIP app, which will effectively, put the first caller on hold.

What is needed is a way to convey to the first caller (PSTN/cellular voice) that the call with the second caller has been put on hold. The VoIP user might not actively put the other person on hold. It can happen whenever the VoIP user receives an incoming voice call during a VoIP call. Usually, the voice call will take priority over any other running app, including the VoIP app, which will effectively, put the other side on hold.

In order to obviate the deficiencies in the art, the disclosed subject matter presents a novel method of providing secondary communication to a first communication device, wherein the first communication device is in primary communication with a second communication device via a network and the first communication device is in communication with the network over a voice channel and the second communication device is in communication with the network over a data channel. The method includes receiving a signaling message from the second communication device when the primary communication is interrupted and substituting the second communication device with a third communication device to establish a secondary communication between the first communication device and the third communication device.

The disclosed subject matter also presents a novel method for alerting the first caller that the second caller has placed the first call between the first caller and the second caller on hold. The method including determining an interruption, has occurred at the second caller's device of the first call; sending a message to a carrier and receiving the message at the carrier. In response to the message, the method includes disconnecting the second caller from the first call; connecting a third party to the first call in place of the second caller; and providing communications from the third party to the first caller.

These and many other objects and advantages of the present subject matter will be readily apparent to one skilled in the art to which the invention pertains from a perusal of the claims, the appended drawings, and the following detailed description of preferred embodiments.

BRIEF DESCRIPTION OF THE DRAWINGS

FIG. **1** is a diagram of a communications environment including various elements which are associated with an Internet protocol (IP) telephony system operating in accordance with the disclosed subject matter;

FIG. **2** is a simplified diagram of a telephony system utilizing an embodiment of the disclosed subject matter.

FIG. **3** is a flow chart for informing a call is placed on hold and then reestablished representing an embodiment of the disclosed subject matter.

FIG. **4** is an illustration of a call flow diagram representing an embodiment of the disclosed subject matter.

FIG. **5** is an illustration of a call flow diagram representing another embodiment of the disclosed subject matter.

FIG. **6** is an exploded diagram of portions of the IP telephony system operating in accordance with an embodiment of the disclosed subject matter.

DETAILED DESCRIPTION OF PREFERRED EMBODIMENTS

The following detailed description of preferred embodiments refers to the accompanying drawings, which illustrate specific embodiments of the invention. Other embodiments having different structures and operations do not depart from the scope of the present invention.

In the following description, the terms VOIP system, VOIP telephony system, IP system and IP telephony system are all intended to refer to a system that connects callers and that delivers data, text and video communications using Internet protocol data communications.

In embodiments of the disclosed subject matter, Session Initiation Protocol (SIP) is used as the connection protocol to execute the call session set up (and subsequent tear down) steps. SIP is a signaling protocol for initiating, managing and terminating media (e.g., voice, data and video) sessions across packet based networks that typically use the Internet Protocol (IP) of which VoIP is an example. The details and functionality of SIP can be found in the Internet Engineering Task Force (IETF) Request for Comments (RFC) Paper No. 3261 entitled, "SIP: Session Initiation Protocol" herein incorporated in its entirety by reference.

Embodiments of the disclosed subject matter also describe the use of Session Description Protocol (SDP) in which participants to a media session offer each other a description of the desired session from the perspective. The details and functionality of SDP can be found in the IETF, RFC Paper No. 3264 entitled "An Offer/Answer Model with the Session Description Protocol (SDP)," which is also incorporated herein in its entirety by reference.

As illustrated in FIG. **1**, a communications environment **100** is provided to facilitate IP enhanced communications. An

IP telephony system **120** enables connection of telephone calls between its own customers and other parties via data communications that pass over a data network **110**. The data network **110** is commonly the Internet, although the IP telephony system **120** may also make use of private data networks. The IP telephony system **120** is connected to the Internet **110**. In addition, the IP telephony system **120** is connected to a publicly switched telephone network (PSTN) **130** via a gateway **122**. The PSTN **130** may also be directly coupled to the Internet **110** through one of its own internal gateways (not shown). Thus, communications may pass back and forth between the IP telephony system **120** and the PSTN **130** through the Internet **110** via a gateway maintained within the PSTN **130**.

The gateway **122** allows users and devices that are connected to the PSTN **130** to connect with users and devices that are reachable through the IP telephony system **120**, and vice versa. In some instances, the gateway **122** would be a part of the IP telephony system **120**. In other instances, the gateway **122** could be maintained by a third party.

Customers of the IP telephony system **120** can place and receive telephone calls using an IP telephone **108** that is connected to the Internet **110**. Such an IP telephone **108** could be connected to an Internet service provider via a wired connection or via a wireless router. In some instances, the IP telephone **108** could utilize the data channel of a cellular telephone system to access the Internet **110**.

Alternatively, a customer could utilize an analog telephone **102** which is connected to the Internet **110** via a telephone adapter **104**. The telephone adapter **104** converts analog signals from the telephone **102** into data signals that pass over the Internet **110**, and vice versa. Analog telephone devices include but are not limited to standard telephones and document imaging devices such as facsimile machines. A configuration using a telephone adapter **104** is common where the analog telephone **102** is located in a residence or business. Other configurations are also possible where multiple analog telephones share access through the same IP adaptor. In those situations, all analog telephones could share the same telephone number, or multiple communication lines (e.g., additional telephone numbers) may provisioned by the IP telephony system **120**.

In addition, a customer could utilize a soft-phone client running on a computer **106** to place and receive IP based telephone calls, and to access other IP telephony systems (not shown). In some instances, the soft-phone client could be assigned its own telephone number. In other instances, the soft-phone client could be associated with a telephone number that is also assigned to an IP telephone **108**, or to a telephone adaptor **104** that is connected to one or more analog telephones **102**.

Users of the IP telephony system **120** are able to access the service from virtually any location where they can connect to the Internet **110**. Thus, a customer could register with an IP telephony system provider in the U.S., and that customer could then use an IP telephone **108** located in a country outside the U.S. to access the services. Likewise, the customer could also utilize a computer outside the U.S. that is running a soft-phone client to access the IP telephony system **120**.

A third party using an analog telephone **132** which is connected to the PSTN **130** may call a customer of the IP telephony system **120**. In this instance, the call is initially connected from the analog telephone **132** to the PSTN **130**, and then from the PSTN **130** through the gateway **122** to the IP telephony system **120**. The IP telephony system **120** then routes the call to the customer's IP telephony device. A third party using a cellular telephone **134** could also place a call to an IP telephony system customer, and the connection would be established in a similar manner, although the first link would involve communications between the cellular telephone **134** and a cellular telephone network. For purposes of this explanation, the cellular telephone network is considered part of the PSTN **130**.

In the following description, references will be made to an "IP telephony device." This term is used to refer to any type of device which is capable of interacting with an IP telephony system to complete an audio or video telephone call or to send and receive text messages, and other forms of communications. An IP telephony device could be an IP telephone, a computer running IP telephony software, a telephone adapter which is itself connected to a normal analog telephone, or some other type of device capable of communicating via data packets. An IP telephony device could also be a cellular telephone or a portable computing device that runs a software application that enables the device to act as an IP telephone. Thus, a single device might be capable of operating as both a cellular telephone and an IP telephone.

The following description will also refer to a mobile telephony device. The term "mobile telephony device" is intended to encompass multiple different types of devices. In some instances, a mobile telephony device could be a cellular telephone. In other instances, a mobile telephony device may be a mobile computing device, such as the Apple iPhone™, that includes both cellular telephone capabilities and a wireless data transceiver that can establish a wireless data connection to a data network. Such a mobile computing device could run appropriate application software to conduct VOIP telephone calls via a wireless data connection. Thus, a mobile computing device, such as an Apple iPhone™, a RIM Blackberry or a comparable device running Google's Android operating system could be a mobile telephony device.

In still other instances, a mobile telephony device may be a device that is not traditionally used as a telephony device, but which includes a wireless data transceiver that can establish a wireless data connection to a data network. Examples of such devices include the Apple iPod Touch™ and the iPad™. Such a device may act as a mobile telephony device once it is configured with appropriate application software.

FIG. **1** illustrates that a mobile computing device with cellular capabilities **136** is capable of establishing a first wireless data connection A, with a first wireless access point **140**, such as a WiFI or WiMax router. The first wireless access point **140** is coupled to the Internet **110**. Thus, the mobile computing device **136** can establish a VOIP telephone call with the IP telephony system **120** via a path through the Internet **110** and the first wireless access point **140**.

FIG. **1** also illustrates that the mobile computing device **136** can establish a second wireless data connection B, with a second wireless access point **142** that is also coupled to the Internet **110**. Further, the mobile computing device **136** can establish a third wireless data connection C, via a data channel provided by a cellular service provider **130** using its cellular telephone capabilities. The mobile computing device **136** could also establish a VOIP telephone call with the IP telephony system **120** via the second wireless connection B or the third wireless connection C.

Although not illustrated in FIG. **1**, the mobile computing device **136** may be capable of establishing a wireless data connection to a data network, such as the Internet **110**, via alternate means. For example, the mobile computing device **136** might link to some other type of wireless interface using an alternate communication protocol, such as the WiMax standard. Throughout the specification, the use of "commu-

nication device(s)" is intended to encompass mobile computing devices **136** as well as cellular and land line devices.

The current subject matter relates to situations where two callers—a first caller communicating over a traditional PSTN or cellular voice channel, and the second caller communicating through a VoIP application over a data channel, are on a call with each other, and the VoIP user receives a second call **208** over a voice channel. More specifically, the subject matter provides a mechanism to alert the first caller that the second caller has put him on hold.

An exemplary scenario in which an embodiment of the disclosed subject matter is advantageously undertaken is shown in FIG. **2**. A first user employing a first communication device **202**, shown as either of a cellular phone or a traditional POTS phone is connected on a first call **206** with the second user employing a second communication device **204**, shown as a mobile computing device.

The first communication device **202** is connected to the first call **206** over a voice channel of either a cellular provider or a Public Switch Telephone Network (PSTN **130**). The PSTN/Cellular Provider **130** interfaces with the IP telephony system **120** via Gateway **122** as described in FIG. **1**. The second communication device **204** connects to the first call **206** over a data channel through a Wi-Fi router **140** and the Internet **110**, as described generally above. As shown in FIG. **2**, during the first call **206**, a second call **208** is initiated over the voice channel (typically a cellular channel) from a third caller **210**. The initiation of the second call **208** during the first call **206** is an Off-Net scenario.

FIG. **3** illustrates an embodiment of a disclosed method **300**. A first call **206** (as described above) is established between the first device **202** and the second device **204** as shown in Block **302**. When a second call **208** is initiated with the second communication device **204**, the first call **206** is interrupted as shown in Block **304**. In most mobile computing devices/smart phones today, a call coming over the voice channel will override all apps, including a VoIP app. The VoIP app will be notified by the operating system of the mobile computing device that it's been moved to the background (which occurs when the call comes in). The VoIP app may then check the telephone's state and the VoIP app recognizes that the second call **208** has come in and in response to the second call, the VoIP app sends a signaling message to the VoIP provider that the first call is now on hold as shown in Block **306**. While FIG. **3** shows the step in Block **306** is performed by the second communication device **204**, the VoIP provider may employ other means, such as detecting the lack of voice packets, to determine the first call **206** has been placed on hold by the second communication device **204**.

The VoIP provider **120** receives the message from the second device as shown in Block **308** and substitutes the second device with a third device (such as device **624** in FIG. **6** and described in greater detail below) to establish a second call **208** between the first communication device and the third device as shown in Block **310**. This substitution switches the media path so that media is now provided from the second caller's end from another source (the third device). The media provided between the third device and the first device is referred to as a secondary communication, and may include a prerecorded message, tone, an interactive menu (press #1 to leave a message, press #2 etc.), or two-way communication with an operator. The third device **624** may be many types of communications equipment that may serve as a source for the secondary communication.

When the interruption is over, the VoIP app recognizes that the call on the voice channel is over and the second caller is ready to re-engage with the first call **206** as shown in Block **314**. If the second call **208** was over a data channel, a solution would be for the VoIP provider to switch back to the first call **206** when it receives a SIP BYE message [however as the second call **208** is over a voice channel, the second communication device **204** sends a message to the VoIP provider as shown in Block **316** to alert the VoIP provider the second call **208** has terminated. The VoIP app may use an API (for example, the API found at http://stackoverflow.com/questions/5028329/ios-4-2-return-to-app-after-phone-call/10635015#10635015 for iOS based devices) to tell the device to re-launch the app once a call has ended.

Upon receiving the message from the second communication device **204** as shown in Block **318**, the VoIP provider **120** restores the connection to the second communication device **204** to the first call **206** and disconnects the third party (third device) as shown in Block **320**.

FIG. **4** is a high level call flow diagram **400** of an embodiment of the disclosed subject matter. During a first call **206** between the second device **204** and the first device **202**, media (voice traffic) **402***a*, **402***b* and **402***c* is transmitted between the second device **204** and a Media relay **622** (see FIG. **6**), the Media relay **622** and the Gateway **122**, and the gateway **122** and the first device **202**. The media is transmitted using Real Time Protocol (RTP) packets or other Voice over Packet (VOP) mechanism on media channels **602** as shown in a system overview **600** as shown in FIG. **6**. For the purposes of FIGS. **4-6**, media channels **602** and signaling channels **603** fall under data channels. When a second call **208** (interruption **470**) over a voice channel comes into the second communication device **204**, the VoIP app will send a SIP message or an API request to the VoIP server **620**. Specifically, the VoIP app resident on the second communication device **204** generates a signaling message **401** to a VoIP server **620** which is part of the IP Telephony System **120** and may comprise proxies and other servers that manage VoIP sessions. If the first call **206** has already been going through a media relay **622** as shown on FIG. **6**, then the VoIP provider can simply have the media relay **622** also handle the secondary communication with the first caller.

The VoIP server **620** generates a message **403** to the Media Relay **622** handling the first call media **402**. The message **403** informs the Media Relay **622** that a change in media source is needed and requests an address for an open port in which to accept media from the new source, or addresses for an open pair of ports. The Media Relay **622** provides a response message **405** to the VoIP server **620**. The response message includes a Session Description Protocol (SDP) which includes the address of the open port upon which the Media Relay **622** expects to receive media and possibly, list of accepted codecs, although not shown in FIG. **4**, SDPs for a new set of inboard/outboard ports may also be provided, as shown in message **505** of FIG. **5**. The port provided by the Media Relay **622** is internally connected to an outboard port that communicates Media **402***b* to the Gateway **122**. In FIG. **6**, for illustration only, the inboard ports (or second caller side ports) are labeled 2 and 3. and the outboard ports (or first caller side ports) are labeled 1 and 4.

Upon receipt of the message **405** from the media relay **622**, the VoIP server **620** generates a message **407** to a third device **624**, which includes the SDP_1 (in FIG. **4**, the message **407** is shown as an INVITE message). The third device **624** may be a Voice Recorder or other type of audio source, for example an online music source, a message recording or playback system, an administrator proxy which can accept user instruction on how to handle the first call (terminate, leave message, call back etc.) or a terminal device associated with a live or simulated operator. In FIG. **6**, two third devices **624***a* and **624***b* are

shown. Third device **624***a* is separate from the Media Relay **622** and is accessed in the same manner as other VoIP endpoints, which is described in FIG. **4**. Third device **624***b* however is associated with a dedicated port on the Media Relay **622** and alternatively may be integral with the Media Relay **622**. The third device **624***b* may be connected directly with outboard port communication with only a message **403** from the VoIP server **620**, in such case the message **403** would preferably include an indication to access the dedicated port and commence the secondary communication. The INVITE **407** includes the SDP provided by the Media Relay **622**, and the third device **624** delivers Media **404***a* to the specified port on the Media Relay **622** which relays the media **404***b* to the Gateway **122** and onto the first device **202** as media **404***c*. Where the Media Relay **622** provides a new set of ports, a re-INVITE message to the Gateway **122** is also required, this re-INVITE message would be similar to the invite message **521** of FIG. **5**.

The second communication device, determines the interruption **470** has concluded, then generates a Message **409** to the VoIP server **620** indicating the second caller is ready to resume the first call communication. The determination that the interruption **470** has concluded may be a function of a VoIP App, or a function handled by the network **120**. In response, the VoIP server generates a Message **411** to the Media Relay **622** indicating a change in media source and requesting the address of a new port on which to receive the media **406***a*. The VoIP server **620** also generates a Bye message **415** to the third Device **624** which terminates the media connection to the Media Relay **622**. The Media Relay **622** replies to the message **411** with message **417** including the SDP with the address of the desired port upon which to receive the Media **406***a*. The VoIP server **620** sends an invite message **419** including the SDP to the second communication device **204**. The second communication device **204** sends and receives media **406***a* to and from the designated port on the Media Relay **622**. The Media **406***a* is then relayed to and from the Gateway **122** as Media **406***b* and to and from the first communication device **202** as Media **406***c*.

FIG. **4**, as discussed above, generally illustrates a scenario in which the outboard port of the Media Relay **622** in communication with the first device remains the same and only the inboard port is changed. FIG. **5**, illustrates not only the method in which a media relay **622** must be selected, but also where both new inboard and outboard ports are provided by the Media relay **622** to the VoIP server **620** for the secondary communication and subsequently for the re-establishment of the primary communication.

If the first call **206** was a direct device to device call, (i.e. not involving a Media relay), the system will send a re-invite to the first caller to put the first communication device in contact with a selected media relay that can handle the secondary communication. FIG. **5** illustrates a high level call flow diagram **500** in which a media relay was not used to handle the first call.

During the first call **206** in the scenario shown in FIG. **5**, between the second device **204** and the first device **202**, media **502***a* and **502***c* is transmitted between the second device **204** and the Gateway **122**, and the Gateway **122** and the first device **202**. When a second call **208** (interruption **570**) over a voice channel comes into the second communication device **204**, the VoIP app resident on the second communication device **204** generates a signaling message **501** to a VoIP server **620**. The VoIP server **620** then selects a Media Relay **622** to handle the communication as shown in Block **555** of FIG. **5**. The selection of the Media Relay **622** may be based on load, proximity, quality of service ("QoS") and/or other characteristics. For example, the VoIP server **620** can select the closest Media relay between the two callers. That selection enables the ability to decrease latency and travel time of the Media **506**.

The VoIP server **620** generates a message **503** to the selected Media Relay **622**. The message **503** informs the Media Relay **622** that a media session needs to be established and requests addresses for open ports on the second callers side and the first callers side. The Media Relay **622** provides a response message **505** to the VoIP server **620**. The response message includes a first SDP_1 and a second SDP_2 which include the addresses of the open ports upon which the Media Relay **622** expects to receive media and send media from, respectively. The ports provided by the Media Relay **622** are internally connected.

Upon receipt of the response message **505** from the Media Relay **622**, the VoIP server **620** generates an Invite **507** to a third device **624** and a Re-Invite **521** to the Gateway **622**. The Invite **507** includes the SDP1 provided by the Gateway **122** to receive media **504***a* and the third device **624** delivers Media **504***a* to the specified port on the Media Relay **622**. The invite **521** includes the SDP2 provided by the Gateway **122** to receive and send media **504***b* from and to the Media Relay **622**, which relays the media **504***b* to the first device **202** as media **504***c*.

The second communication device through the VoIP app upon determining the interruption **570** has concluded, generates a Message **509** to the VoIP server **620** indicating the second caller is ready to resume the first call communication. In response, the VoIP server generates a Message **511** to the Media Relay **622** indicating a change in media source and requesting the address of a new ports on which to receive the media **506***a* and **506***b*. The VoIP server **620** also generates a Bye message **515** to the third Device **624** which terminates the media connection to the Media Relay **622**. The Media Relay **622** replies to the message **511** with message **517** including a new SDP1* and SDP2* which include the address of the desired ports upon which to receive and send the Media **506***a* and **506***b* (the * indicating the ports to be used to re-establish the primary communication as distinguished from the ports used to establish the secondary communication). The VoIP server **620** sends an invite message **519** including the SDP1* to the second communication device **204**, and a re-invite message **523** to Gateway **122** including SDP2*. The second communication device **204** sends and receives media **506***a* to and from the designated port on the Media Relay **622**. The Media **506***a* is then relayed to and from the Gateway **122** as Media **506***b* and to and from the first communication device **202** as Media **506***c*. The ports selected to receive the Media **504***a* from the third device **624** and the Media **506***a* from the second device **204**, may be different or the same. If they are the same ports, the Bye message **515** must be sent and a response acknowledgement received prior to sending invite **519**.

Other messages, such as authentication, acknowledgements and responses that are not necessary for the understanding of the disclosed subject matter are not shown in FIGS. **4** and **5**, for example (ack, 200 Ok, trying, challenge, etc.)

While SIP messaging nomenclature is used in describing the VoIP signaling, other signaling, such as Sigtran, SS7, H.323, XMPP and ISDN are also envisioned.

While preferred embodiments of the present subject matter have been described, it is to be understood that the embodiments described are illustrative only and that the scope of the invention is to be defined solely by the appended claims when

accorded a full range of equivalence, many variations and modifications naturally occurring to those of skill in the art from a perusal hereof.

What we claim is:

1. A method of providing secondary communication to a first communication device, wherein the first communication device is in primary communication with a second communication device via a network interface, and the first communication device connected to the network interface over a voice channel and the second communication device connected to the network interface over a data channel via VoIP, comprising:

receiving a signaling message from the second communication device indicative of a determined interruption to the primary communication,

substituting the second device with a third communication device in response to the signaling message, to thereby establish the secondary communication between the first communication device and the third communication device; wherein the primary communication and the secondary communication is communication of media, wherein the step of substituting the second device with a third communication device further comprising:

sending a third signaling message to a media relay

in response to the third signaling message, establishing communications between the media relay and the third communications device.

2. The method of claim **1**, wherein the step of establishing communications between the media relay and the third communications device further comprises:

sending an invite to the third communications device, wherein the invite includes an address to a third port of the media relay, wherein the secondary communication from the third communications device are received at the third port of the media relay.

3. The method of claim **2**, comprising; connecting the third port of the media relay with a first port of the media relay, wherein the first port is connected via the network interface with the first communication device.

4. The method of claim **3**, wherein the third communication device is an audio messaging device.

5. A method of providing secondary communication to a first communication device, wherein the first communication device is in primary communication with a second communication device via a network interface, and the first communication device connected to the network interface over a voice channel and the second communication device connected to the network interface over a data channel via VoIP, comprising:

receiving a signaling message from the second communication device indicative of a determined interruption to the primary communication,

substituting the second device with a third communication device in response to the signaling message, to thereby establish the secondary communication between the first communication device and the third communication device; wherein the primary communication and the secondary communication is communication of media, wherein the step of substituting the second device with a third communication device further comprising:

selecting a media relay;

sending a third signaling message to the media relay;

establishing communications between the media relay and the third communications device and establishing communications between the media relay and the first communication device via the network interface.

6. The method of claim **5**, wherein the step of establishing communications between the media relay and the first communication device comprises:

substituting the second device with a third communication device further comprising:

sending an invite to the network interface, wherein the invite includes an address of a first port of the media relay.

7. The method of claim **6**, wherein the step of establishing communications between the media relay and the third communications device further comprises:

sending a second invite to the third communications device, wherein the second invite includes an address to a third port of the media relay, wherein the secondary communication from the third communications device are received at the third port of the media relay.

8. The method of claim **7**, comprising; connecting the third port of the media relay with the first port of the media relay, to establish the secondary communication between the third communication device and the first communication device via the network interface and media relay.

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